CLAIMS

1. (currently amended) A method for using a computer processor to interpolatively code ~~input~~ a digitized audio waveform input ~~signals,~~ signal having a first bitrate into a coded audio waveform output signal having a second bitrate lower than said first bitrate, said method comprising the steps of:

extracting ~~in which said signals decomposed into or are composed of~~ a slowly evolving waveform from the digitized audio waveform input signal;

estimating a dispersion phase of an excitation signal; ~~comprising inputting waveform signals to the computer and~~

locking onto a most probable pitch period;

quantizing a sequence of gain trajectory correlation values;

using the computer processor to transform the extracted slowly evolving waveform, the estimated dispersion phase, the most probable pitch period and the quantized sequence of gain trajectory values into an interpolatively coded audio waveform output signal with said lower bitrate; and

outputting said coded audio waveform output signal,

wherein said method comprises using the computer processor to execute at least one ~~a~~ step selected from the group consisting of:

(a) ~~using the computer processor to perform~~ performing an analysis-by-synthesis vector quantization of the dispersion phase such that ~~the~~ a linear shift phase ~~attribute~~ residual is ~~reduced or eliminated from the quantization~~ minimized;

(b) ~~using the computer processor to process~~ computing a weighted average of a group of adjacent pitch values ~~and weighting them to compute a weighted average~~ in order to compute the most probable ~~value of~~ pitch period;

(c) ~~using the computer processor to incorporate~~ performing spectral and temporal pitch searching in order to compute the most probable pitch period, such that the temporal pitch searching is performed at a different rate than the spectral pitch searching;

(d) ~~using the computer processor to incorporate~~ incorporating temporal weighting in

~~the~~ an analysis-by-synthesis vector-quantization of the gain ~~sequence~~ trajectory correlation values;

(e) ~~using the computer processor to quantize~~ quantizing adjacent gain trajectory correlation values by analysis-by-synthesis vector-quantization without downsampling or interpolation ~~of the gain values~~;

(f) ~~using the computer processor to incorporate~~ incorporating switched prediction ~~or switched~~ filtering in an analysis-by-synthesis vector-quantization of the sequence of gain ~~sequence~~ trajectory correlation values;

~~(g) using a coder in which a plurality of bits therein are allocated to the vector-quantization of the dispersion phase of the slowly evolving waveform phase from which the linear shift attribute was reduced or removed; and~~

~~(h)~~ (g) ~~using the computer processor for~~ temporal pitch searching with ~~using~~ varying segment boundaries ~~of the summations used in computing the similarity or an equivalent distortion measure used for the pitch search~~.

2. (currently amended) The method of claim 3 ~~1~~ in which said ~~signal~~ digitized audio waveform input signal is representative of speech and said coded output signal has a subjective speech quality at 4 kbps better than that of G.723 coding at 6.3 kbps.

3. (currently amended) The method of claim 1 in which said method incorporates all of ~~a plurality of~~ steps (a) through (g) ~~(h)~~.

4. (currently amended) The method of claim 1 ~~further comprising the step of analysis-by-synthesis vector-quantization of the slowly evolving waveform~~, wherein distortion is reduced ~~in the signal~~ by obtaining ~~the~~ an accumulated weighted distortion between a sequence of input waveforms and a sequence of quantized and interpolated waveforms.

5. (currently amended) The method of claim 1 wherein said at least one step is step (a) further comprising providing at least one codebook containing magnitude and dispersion phase information for predetermined waveforms, ~~and in which the step of analysis-by-synthesis quantization of the dispersion phase comprises~~ approximately

aligning a ~~the~~ linear phase ~~of one or the other of the~~ input or ~~and~~ output, then iteratively shifting the approximately aligned linear phase input or output, comparing the shifted input or output to a plurality of waveforms reconstructed from the magnitude and dispersion phase information contained in said at least one codebook, and selecting the reconstructed waveform that best matches one of the iteratively shifted inputs or outputs.

6. (currently amended) The method of claim 1 ~~in which in the method of temporal pitch searching of~~ wherein said at least one step includes step ~~(d)~~ (g) ~~comprises defining~~ and said varying segment boundaries ~~of segments of said summations~~ are used to compute ~~similarity selecting the~~ a best boundary ~~such that maximizing the similarity or minimizing the distortion measure~~ by iteratively shifting and changing the length of the segments ~~used for the summations used in the measure computations~~.

7. (currently amended) The method of claim 1 ~~in which~~ wherein said at least one step is step (c), the spectral pitch search is conducted at a first rate and the temporal pitch ~~searches are~~ searching is conducted at a second rate different from said first rate.

8. (currently amended) The method of claim 1 ~~in which the~~ wherein said at least one step is step (d) ~~of the~~ and said temporal weighting ~~in the analysis-by-synthesis vector-quantization of the signal gain sequence is changed as a function of time whereby to emphasize~~ emphasizes local high energy events in the input signal.

9. (currently amended) The method of claim 1, ~~further comprising~~ wherein said at least one step is step (e) or step (f) and ~~applying~~ both high correlation and low correlation synthesis filters are applied to a vector quantizer codebook ~~in the analysis by synthesis vector quantization of the signal gain whereby to add self correlation to the codebook vectors, in which selection between~~ and a selected one of the high and low correlation synthesis filters ~~is made to maximize~~ maximizes similarity between ~~the~~ an input target gain vector and a reconstructed vector.

10. (canceled)

11. (canceled)

12. (currently amended) A method for using a computer to quantize audio waveforms comprising:

inputting digitized audio waveform signals to the computer,

using the computer to generate a plurality of adjacent quantized and interpolated output waveforms having a lower bitrate than the input waveform signals; ~~and~~

~~determining~~ using the computer to determine ~~from said signals~~ an accumulated distortion between ~~adjacent~~ the input ~~waveforms~~ waveform signals and each of said adjacent quantized and interpolated output waveforms; and

generating a reconstructed waveform using said accumulated distortion.

13. (currently amended) A method for using a computer to interpolatively code ~~input~~ digitized audio waveform signals comprising:

inputting the digitized audio waveform signals to the computer,

~~determining said~~ extracting a slowly evolving waveform from said signals;

extracting a dispersion phase from said slowly evolving waveform; ~~and~~

performing an analysis-by-synthesis quantization of ~~the~~ said dispersion phase; and

using the quantized dispersion phase to transform the input waveform signals into an interpolatively coded output waveform signals having a lower bitrate than said input waveform signals.

14. (currently amended) The method of claim 13 further comprising:

providing at least one codebook containing magnitude and dispersion phase information for predetermined waveforms,

approximately aligning ~~the~~ a linear phase of the ~~input~~ digitized audio waveform signals,

then iteratively shifting ~~said~~ the approximately aligned linear phase ~~input, and/or comparing the shifted input, or equivalently shifting the quantized vector,~~ relative to a plurality of vectors reconstructed from the magnitude and

dispersion phase information contained in said at least one codebook, and selecting one of the thus reconstructed ~~vector~~ vectors that best matches ~~the input vector or~~ one of the iteratively shifted input vectors.

15. (currently amended) A method for using a computer processor to interpolatively code an ~~input~~ audio waveform having certain ~~signals in which the signal decomposed into or composed of~~ attributes and ~~or~~ components including ~~one of which is~~ a slowly evolving waveform~~, which has or from which one can extract~~ and an associated dispersion phase, comprising:

inputting digitized audio waveform signals to the computer processor and ~~incorporating~~ using the computer to perform analysis-by-synthesis quantization of the associated dispersion phase, including

providing at least one codebook containing magnitude and dispersion phase information for predetermined waveforms,

crudely aligning a linear phase of the input vector, then iteratively shifting said crudely aligned linear phase input vector relative ~~, and/or comparing the iteratively shifted input vector, or equivalently shifting a quantized vector,~~ to a plurality of vectors reconstructed from the magnitude and dispersion phase information contained in said at least one codebook, and

selecting the reconstructed vector that best matches the input vector ~~or one of the iteratively shifted input vectors~~, in which a distortion measure for a given data vector is determined by a perceptually weighted average of distortion measures for harmonics of the given data vector, wherein the perceptual weighted average ~~weighting~~ combines a spectral-weighting and synthesis ~~and~~ in which an average global distortion measure for a particular vector set M is an average of distortion measures for the ~~data~~ vectors in M and ~~including the step of minimizing the~~ global distortion ~~thereof~~ is minimized by using a centroid formula to determine phases of harmonics; and

using the thus selected best matching reconstructed vector to transform the input

waveform signals into interpolatively coded output waveform signals having a lower bitrate than said input waveform signals.

16. (currently amended) The method of claim 15, wherein the centroid formula uses both input waveform coefficients and quantized slowly evolving waveform coefficients.

17. (currently amended) A method for using a computer to interpolatively code digitized audio ~~input~~ waveform signals, comprising:

inputting the digitized audio waveform signals to the computer

performing spectral pitch searching ~~and temporal pitch searches~~ searching on said signals,

performing temporal pitch searching on said signals;

determining a number of adjacent pitch ~~values, and~~ values;

computing a most probable pitch value by computing ~~the~~ a weighted average pitch value from the adjacent pitch values; and

using the thus computed most probable pitch value to transform the input waveform signals into interpolatively coded output waveform signals having a lower bitrate than said input waveform signals..

18. (currently amended) The method of claim 17 in which in the ~~method~~ step of performing ~~searching the~~ temporal domain pitch searching comprises

defining a boundary for a segment used for ~~the~~ summations in a ~~the~~ computed measure used for the pitch searching, ~~search,~~ and

selecting the boundaries of the segment that ~~maximize the similarity or minimize~~ optimizes the ~~distortion~~ computed measure ~~used for the pitch search,~~ by iteratively shifting and expanding the segment.

19. (currently amended) A method for using a computer to interpolatively code digitized audio ~~input~~ waveform signals comprising:

inputting the digitized audio waveform signals to the computer,

performing spectral domain and temporal domain pitch searches to lock onto a most

probable pitch period of each of the~~-signal~~ signals,

determining a number of adjacent pitch values, ~~and~~

then computing the most probable pitch value by computing a weighted average pitch ~~value~~ value, and

using the thus computed most probable pitch value to transform the digitized audio waveform signals into interpolatively coded output waveform signals having a lower bitrate than said digitized audio waveform signals.

~~in which~~ wherein ~~the method of searching~~ the temporal domain pitch searching is based on harmonic matching using varying segment boundaries.

20. (currently amended) A method of using a computer to interpolatively code digitized audio waveform input ~~waveform~~ signals comprising

inputting the digitized audio waveform signals to a computer;

using a weighted average using normalized correlations for weights to compute ~~one~~ a weighted average pitch value out of a set of pitch values of the waveform ~~signal~~ signals, ~~; and using the~~ wherein each of the pitch values is used to regenerate a respective reconstructed waveform; and

using the thus computed weighted average pitch value to transform a digitized audio waveform signal into an interpolatively coded output waveform signal having a lower bitrate than said digitized audio waveform signals.

21. (currently amended) The method of claim 19 in which the ~~performing~~ spectral domain ~~pitch~~ and temporal domain pitch searches are conducted respectively at 100 Hz and 500 Hz.

22. (currently amended) A method for using a computer to interpolatively code digitized audio ~~input~~ waveform signals , comprising:

inputting the digitized audio waveform signals to the computer; ~~and~~

performing analysis-by-synthesis vector quantization of ~~the~~ a gain sequence of each of the waveform input signals, and regenerating an output signal using said gain sequence; and

using the resultant vector quantized gain sequence value to transform a digitized

audio waveform signal into an interpolatively coded output waveform signal having a lower bitrate than said digitized audio waveform signals..

23. (currently amended) The method of claim 22 including using ~~temporal weighting, and in which the~~ temporal weighting which is changed as a function of time whereby to emphasize local high energy events in the input ~~signal~~ signals.

24. (currently amended) The method of claim 23, further comprising applying a synthesis filter or predictor, which introduces selected ~~high~~ correlation ~~or low correlation~~ to a vector quantizer codebook in the analysis-by-synthesis vector-quantization of the signal gain sequence to add selected self correlation to the codebook vectors.

25. (previously presented) The method of claim 24 in which selection between the high and low correlation synthesis filters or predictor is made to maximize similarity between signal and reconstructed vectors.

26. (previously presented) The method of claim 22, comprising using each value of gain index in the analysis-by-synthesis vector-quantization of the signal gain.

27. (previously presented) The method of claim 22 wherein each value of gain index is used to select from a plurality of shapes and associated predictors or filters, each of which is used to generate an output shape vector, and comparing the output shape vector to an input shape vector.

28. (previously presented) The method of claim 27 in which said plurality of shapes has a predetermined number of values in the range of 2 to 50.

29. (previously presented) The method of claim 27 in which said plurality of shapes has a predetermined number of values in the range of 5 to 20.

30. (currently amended) A method for using a computer to interpolatively code ~~input~~ audio waveform signals, comprising:

inputting a digitized waveform ~~signals~~ signal to the computer,

decomposing said signal into a slowly evolving waveform, ~~and~~

~~using a coder in which a plurality of bits therein are allocated to the~~ performing a vector-quantization of a dispersion phase of the slowly evolving waveform ~~phase~~ from which ~~the~~ a linear shift attribute was reduced or removed and

transforming the digitized audio waveform signals into interpolatively coded output waveform signals having a lower bitrate than said digitized audio waveform signals, wherein a plurality of bits of the coded output waveform signals are allocated to the vector-quantized dispersion phase with the reduced linear shift attribute.

31. (previously presented) The method of claim 30 in which at least one bit is allocated to the dispersion phase.

32. (currently amended) A method for using a computer to interpolatively code audio ~~input~~ waveform signals comprising:

inputting digitized audio waveform signals to a computer; ~~and~~

using at least one processor of the computer to:

determine input vectors representing the waveform signals;

determine interpolated vectors for modeling the input vectors;

compute an accumulated weighted distortion between the input vectors and the interpolated vectors as a sum of a modeling distortion and a quantization distortion; and

determine an optimal vector which minimizes the modeling distortion; and

using the thus computed accumulated weighted distortion to transform the digitized audio waveform signals into interpolatively coded output signals having a lower bitrate than said digitized audio waveform signals..

33. (currently amended) The method of claim 32 further comprising:

using at least one processor of the computer to determine a respective quantized vector ~~using~~ from the optimal vector.

34. (currently amended) The method of claim 17 in which the step of computing a number of adjacent pitch values includes ~~some weight associated with their~~

~~probability , and~~ using a respective function of normalized autocorrelations obtained for each pitch value~~, or some function of autocorrelation,~~ as ~~its~~ an associated probability weight ~~used~~ to compute the weighted average pitch value.

35. (previously presented)The method of claim 12 including using accumulated spectrally weighted distortion.

36. (previously presented) The method of claim 22 including using a switch predictive synthesis filter or predictor.